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| EXAMINER | |
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| WOZNIAK, JAMES S | |

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**Please find below and/or attached an Office communication concerning this application or proceeding.**

The time period for reply, if any, is set in the attached communication.

| Office Action Summary | Application No. | Applicant(s) | |
|---|---|---|---|
| | 10/019,617 | CHANDRAN ET AL. | |
| | Examiner | Art Unit | |
| | James S. Wozniak | 2626 | |

*-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --*

**Period for Reply**

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

**Status**

1)☒ Responsive to communication(s) filed on 06 August 2007.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

**Disposition of Claims**

4)☒ Claim(s) 1,3-34 and 36-68 is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) 1,3-34 and 36-68 is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

**Application Papers**

9)☐ The specification is objected to by the Examiner.

10)☒ The drawing(s) filed on 28 May 2002 is/are: a)☐ accepted or b)☒ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

**Priority under 35 U.S.C. § 119**

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

**Attachment(s)**

1) ☒ Notice of References Cited (PTO-892)
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☐ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date _____.
4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .
5) ☐ Notice of Informal Patent Application
6) ☐ Other: _____.

## DETAILED ACTION

### *Response to Amendment*

1. In response to the office action from 2/5/2007, the applicant has submitted a request for continued examination, filed 8/6/2007, amending independent claims 1, 29, 34, and 62, while adding claims 67-68 and arguing to traverse the art rejection based on the limitations regarding adjusting parameters in a presence of speech, noise, and a combination thereof *(Amendment, Pages 21-22)* and the use of a hybrid coded-linear format *(Amendment, Pages 22-23)*. Applicant's arguments have been fully considered, however the previous rejection *(from 6/12/2006)* is maintained due to the reasons listed below in the response to arguments.

2. In response to the previous 35 U.S.C. 101 rejection the applicants argue that claim 62 now features a step for "transmitting the first and second bits in an adjusted state to a far end device to produce a corresponding audible signal with managed echo for an end user", which overcomes the corresponding 35 U.S.C. 101 rejection *(Amendment, Page 21)*. The examiner notes that the transmission of such bits to a far end device to present the bits in audio form to a user would be a "useful, concrete, and tangible result", and thus, the aforementioned rejection has been withdrawn. Such claim language, however, raises a claim objection due to the inclusion of "managed echo" because the claim is directed towards managing a noise and no near end/far end audio is received or processed.

## *Response to Arguments*

3. Applicant's arguments have been fully considered but they are not persuasive for the following reasons:

The applicants have amended claims 1, 29, and 62 to revert them to their previous format and argues that these claims would not be single means claims because they do not utilize the "means for" or "step for" language *(amendment, Pages 20).* First, it is noted that Claims 29 and 62 are not a single means/step claims because they have been further amended to include the additional step/module for "transmitting". In addressing the applicants' arguments regarding claim 1, the examiner points out that although the aforementioned claims do not expressly utilize "means for" language, they are nonetheless means-plus-function claims because the "processor" of claim 1 is described in terms of the function it performs *(Seal-Flex, Inc. v. Athletic Track and Court Construction, 172 F.3d 836, 850, 50 USPQ2d 1225, 1234 (Fed. Cir.1999) (Radar, J., concurring) ("claim elements without express step-plus-function language may nevertheless fall within 112 6 if they merely claim the underlying function without recitation of acts for performing that function)) (also see MPEP 2181).* Claim 1 recites a "processor" (means) to (for) "read...generate...replace" (function), thus, claim 1 is in means-plus-function format and accordingly is a single means claim.

Now addressing the art rejections, the applicants first argue that Yue et al *(U.S. Patent: 6,026,356)* fails to teach the added limitation "a processor...to generate an adjusted first parameter in a presence of speech, noise, and combination thereof" because Yue reduces/eliminates artifacts that distort the acoustic background noise and outputs detected

speech data *(Amendment, Pages 21-22)*. Upon considering the applicants arguments with respect to claim 1, the examiner notes that it appears the applicants are arguing that the first audio parameter recited in claim 1 comprises a combination of speech and noise *(i.e., noisy speech)*, the latter of which is managed by adjusting and replacing the first parameter. While the examiner notes that the language of new claim 67 would seem to support this argument, claim 1 recites only that "a processor...generate an adjusted first parameter in a presence of speech, noise, and combination thereof". Given the broadest reasonable interpretation of the claim, such a limitation can be read as the processor performing its function *(generating an adjusted first parameter)* in a noisy speech environment *(in a presence of speech, noise, and combination thereof)*. As such, it is noted that Yue recites speech coding parameter replacement implemented in a mobile telephone environment *(Col. 1, Lines 50-56; and Col. 5, Line 66- Col. 8, Line 11)* in which speech is subjected to distorting background noise *(i.e., noisy speech)*. Thus, due to the current broad claim scope, the noisy mobile telephone environment taught by Yue anticipates the added limitation because his processing means performs its functions in said environment.

The applicants traverse the art rejection of claim 34 for reasons similar to claim 1 *(Amendment, Page 22)*. In regards to such arguments, see the response directed to claim 1.

The applicants traverse the art rejection of dependent claims 3-28 and 36-61 for reasons similar to claim 1 *(Amendment, Pages 22-24)*. In regards to such arguments, see the response directed to claim 1.

With respect to Claims 29 and 62, the applicants argue that Yue fails to teach a network with a hybrid coded/linear format *(Amendment, Pages 22-23)*. In response, the examiner notes that such a coding format is well known in the prior art as is evidenced by the applicant's

admitted prior art *(Background of the Invention, Pages 4-5 and Fig. 3)*. As such, the applicants' arguments have been fully considered, but are moot with respect to the new grounds of rejection in view of the applicant's admitted prior art.

The applicants traverse the art rejection of dependent claims 30-33 and 63-66 for reasons similar to claims 29 and 62 *(Amendment, Pages 23-24)*. In regards to such arguments, see the response directed to claims 29 and 62.

### *Drawings*

4. **Figures 1-7** should be designated by a legend such as --Prior Art-- because only that which is old is illustrated *(see Background of the Invention, Pages 1-15)*. See MPEP § 608.02(g). Corrected drawings in compliance with 37 CFR 1.121(d) are required in reply to the Office action to avoid abandonment of the application. The replacement sheet(s) should be labeled "Replacement Sheet" in the page header (as per 37 CFR 1.84(c)) so as not to obstruct any portion of the drawing figures. If the changes are not accepted by the examiner, the applicant will be notified and informed of any required corrective action in the next Office action. The objection to the drawings will not be held in abeyance.

### *Claim Objections*

5. Claims are objected to because of the following informalities:

**Claims 29 and 62** contain the limitation "audible signal with managed echo". It is believed that this limitation should read –audible signal with a managed noise characteristic-- because the preambles note that the method/system is for "managing the noise characteristic", the claimed adjustment is performed to manage the noise characteristic, and the specification does not appear to mention how an adjusted noise characteristic of a speech parameter removes echo or receiving/processing near end and far end speech signals *(Fig. 33)*. In light of the above, it is believed that "audible signal with managed echo" should read –audible signal with a managed noise characteristic--. Also, this limitation will be considered as a managed noise for the application of the prior art of record.

Dependent claims 30-33 and 63-66 fail further limit objected independent claims 29 and 62, and thus, are also objected to by virtue of their dependency.

Appropriate correction is required.

## *Claim Rejections - 35 USC § 112*

6. The following is a quotation of the first paragraph of 35 U.S.C. 112:

> The specification shall contain a written description of the invention, and of the manner and process of making and using it, in such full, clear, concise, and exact terms as to enable any person skilled in the art to which it pertains, or with which it is most nearly connected, to make and use the same and shall set forth the best mode contemplated by the inventor of carrying out his invention.

7. **Claims 1 and 3-28** are rejected under 35 U.S.C. 112, first paragraph, as failing to comply with the enablement requirement. The claim(s) contains subject matter that was not described in the specification in such a way as to enable one skilled in the art to which it pertains, or with which it is most nearly connected, to make and/or use the invention.

Claim 1 recites "a processor responsive to..." but lack means for performing the operations that the processor performs.

A single means claim, i.e., where a means recitation does not appear in combination with another recited element of means, is subject to an undue breadth rejection under 35 U.S.C. 112, first paragraph. In re Hyatt, 708 F.2d 712, 714-715, 218 USPQ 195, 197 (Fed. Cir. 1983) (A single means claim which covered every conceivable means for achieving the stated purpose was held nonenabling for the scope of the claim because the specification disclosed at most only those means known to the inventor.). When claims depend on a recited property, a fact situation comparable to Hyatt is possible, where the claim covers every conceivable structure (means) for achieving the stated property (result) while the specification discloses at most only those known to the inventor.

Dependent claims 3-28 do not remedy the lack of enablement issue noted above with respect to claim 1, and therefore, are also rejected under 35 U.S.C. 112, first paragraph, as failing to comply with the enablement requirement.

***Claim Rejections - 35 USC § 102***

8. The following is a quotation of the appropriate paragraphs of 35 U.S.C. 102 that form the basis for the rejections under this section made in this Office action:

> A person shall be entitled to a patent unless –
>
> (e) the invention was described in (1) an application for patent, published under section 122(b), by another filed in the United States before the invention by the applicant for patent or (2) a patent granted on an application for patent by another filed in the United States before the invention by the applicant for patent, except that an international application filed under the treaty defined in section 351(a) shall have the effects for purposes of this subsection of an application filed in the United States only if the international application designated the United States and was published under Article 21(2) of such treaty in the English language.

9. **Claims 1, 3, 13-15, 16-18, 20-21, 25-28, 34, 36, 46-48, 49-51, 53-54, and 58-61** are rejected under 35 U.S.C. 102(e) as being anticipated by Yue et al *(U.S. Patent: 6,026,356)*.

With respect to **Claims 1 and 34**, Yue discloses:

A processor responsive to the compression code of the digital signals to read at least a first parameter, wherein the reading includes partially decoding the first parameter *(analyzing a compressed speech data frame and parameter extraction, Col. 5, Lines 50-Col. 6, Line 53)*;

Responsive to the compression code and the first parameter, generating an adjusted first parameter in a presence of speech, noise, and combination thereof and replacing the first parameter with the adjusted first parameter *(generating and substituting new LPC coefficients, Col. 5, Line 66- Col. 8, Line 11; and parameter replacement implemented in a mobile telephone (i.e., noisy speech) environment, Col. 1, Lines 50-56)*.

With respect to **Claims 3 and 36**, Yue discloses:

The first parameter comprises codebook gain, and wherein the processor modifies the codebook gain to modify the codebook vector contribution to the noise characteristic *(codebook gain and gain correction, Col. 3, Lines 47-67; and Col. 4, Lines 47-67)*.

With respect to **Claims 13 and 46**, Yue discloses:

The plurality of parameters comprises pitch gain wherein the plurality of parameters further comprises a codebook gain, wherein said processor comprises a pitch synthesis filter, wherein said processor performs said plurality of decoding steps to generate a first vector, wherein said processor scales said first vector by said codebook gain to generate a scaled codebook vector, wherein said processor filters said scaled codebook vector through said pitch synthesis filter to generate a second vector, wherein said processor generates a power signal

representing the power of said second vector, wherein said processor is responsive to said pitch gain and said power signal to generate said adjusted first parameter, and wherein said adjusted first parameter comprises an adjusted pitch gain *(excitation parameters comprising codebook gain and LPC coefficients, which are processed using a synthesis filter to obtain information representing power and modified according to a correction factor, Col. 3, Line 36- Col. 4, Line 15; and Col. 6, Line 30- Col. 8, Line 11).*

With respect to **Claims 14 and 47**, Yue further discloses the use of LPC excitation parameters corresponding to entries in a codebook *(Col. 3, Line 36- Col. 4, Line 14).*

With respect to **Claims 15 and 48**, Yue discloses:

The first parameter comprises a codebook vector comprising pulses using variable sets of amplitudes, wherein said processor analyzes said sets to identify the powers of said noise characteristic represented by said sets, wherein said processor identifies a first set representing a power less than the power represented by said sets other than said first set, and wherein said processor adjusts said pulses according to said first set to generate said adjusted parameter *(detecting noise coefficient sets which have a lower power than speech coefficients and using the coefficients in adjusting LPC coefficients, Col. 5, Line 66- Col. 8, Line 11; and Col. 3, Line 36- Col. 4, Line 14).*

With respect to **Claims 16 and 49**, Yue discloses:

The plurality of decoding steps further comprises at least one decoding step that does not substantially affect the management of the noise characteristic and wherein the processor avoids performing the at least one decoding step *(avoiding decompression processing, Col. 8, Lines 36-46).*

With respect to **Claims 17 and 50**, Yue discloses:

The one decoding step comprises post-filtering *(avoiding synthesis processing, Col. 5, Lines 50-65).*

With respect to **Claims 18 and 51**, Yue discloses:

The compression comprises a linear predictive code *(Col. 4, Lines 9-15).*

With respect to **Claims 20 and 53**, Yue discloses:

The compression code comprises code-excited linear prediction code *(CELP, Col. 1, Lines 18-32).*

With respect to **Claims 21 and 54**, Yue discloses:

The first parameter is a quantized first parameter and wherein the processor generates the adjusted parameter in part by quantizing the adjusted first parameter before replacing the first parameter with the adjusted first parameter *(quantizing correction bits before re-insertion into a data frame, Col. 4, Lines 47-67).*

With respect to **Claims 25 and 58**, Yue discloses:

The processor is responsive to the compression code to perform at least one of a plurality of the decoding steps to generate decoded signals and wherein the processor is responsive to the decoded signals and the first parameter to generate the adjusted first parameter *(extracting LPC coefficients from a compressed speech signal for noise modification, Col. 5, Line 66- Col. 8, Line 11; and Fig. 4).*

With respect to **Claims 26 and 59**, Yue discloses:

The first parameter is selected from the group consisting of: codebook vector, codebook gain, pitch gain, and LPC coefficient representations, including line spectral frequencies and log area ratios *(Col. 3, Line 36- Col. 4, Line 14)*.

With respect to **Claims 27 and 60**, Yue discloses:

The first parameter comprises a representation of LPC coefficients, wherein the processor is responsive to the compression code and the representation to determine the spectral regions affected by noise and to generate the adjusted first parameter to manage the noise characteristic in those regions and wherein the adjusted first parameter comprises an adjusted representation of LPC coefficients *(noise adjustment of LPC coefficients utilizing speech detection, Col. 5, Line 66- Col. 8, Line 11)*.

With respect to **Claims 28 and 61**, Yue discloses:

The representation of LPC coefficients is selected from the group consisting of line spectral frequencies and log area ratios *(Col. 4, Lines 9-15)*.

***Claim Rejections - 35 USC § 103***

10. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

11. **Claims 4-5, 22-24, 37-38, and 55-57** are rejected under 35 U.S.C. 103(a) as being unpatentable over Yue et al in view of Swaminathan *(U.S. Patent: 5,495,555)*.

With respect to **Claims 4 and 37**, Yue teaches the system for reducing noise in compressed speech data by substituting adjusted speech coefficients, as applied to Claims 1 and 34. Yue does not specifically suggest substituting speech coefficients based upon pitch gain, codebook gain, and signal to noise ratio (SNR), however Swaminathan discloses a means for selecting an optimal codebook gain based upon a pitch gain, codebook gain, and a SNR *(Col. 12, Lines 40-67)*.

Yue and Swaminathan are analogous art because they are from a similar field of endeavor in speech coding. Thus, it would have been obvious to a person of ordinary skill in the art, at the time of invention, to modify the teachings of Yue with the optimal codebook gain selection means disclosed by Swaminathan in order to achieve effective speech coding processing for voiced to unvoiced transitions *(Swaminathan, Col. 3, Lines 31-41)*.

With respect to **Claims 5 and 38**, Swaminathan further discloses:

The signal to noise ratio comprises a ratio involving noisy signal power and noise power of the audio signal *(SNR, which is a ratio of signal to noise power, Col. 12, Lines 40-67)*.

With respect to **Claims 22 and 55**, Yue teaches the system for reducing noise in compressed speech frames by substituting adjusted speech coefficients, as applied to Claims 1 and 34, while Swaminathan discloses subframe-based processing *(Col. 2, Line 60- Col. 7, Line 41)*.

With respect to **Claims 23 and 56**, Yue further discloses the frame-by-frame noise processing method and system as shown in Fig. 4.

**Claims 24 and 57** contain subject matter similar to Claims 22 and 55, and thus, are rejected for the same reasons. Also, Yue further discloses generating substitute LPC coefficients based on past speech frame coefficients *(Col. 6, Lines 1-28).*

12. **Claims 6, 9, 39, and 42** are rejected under 35 U.S.C. 103(a) as being unpatentable over Yue et al in view of Oshikiri et al *(U.S. Patent: 5,878,387).*

With respect to **Claims 6 and 39**, Yue teaches the system for reducing noise in compressed speech data by substituting adjusted speech coefficients, as applied to Claims 1 and 34. Yue does not specifically suggest the use of a codebook gain and pitch gain at a buffer as recited in claims 6 and 39, however Oshikiri discloses the use of such gain factors *(codebook gain and optimal pitch gain selection, Col. 11, Lines 1-8; and Col. 12, Line 1- Col. 13, Line 11).*

Yue and Oshikiri are analogous art because they are from a similar field of endeavor in speech coding. Thus, it would have been obvious to a person of ordinary skill in the art, at the time of invention, to modify the teachings of Yue with the use of a codebook gain and pitch gain at a buffer as disclosed by Oshikiri in order to provide a means for obtaining sufficient pitch information for high quality voice reproduction at a decoder *(Oshikiri, Col. 4, Lines 12-17).*

With respect to **Claims 9 and 42**, Oshikiri further discloses:

The first parameter comprises pitch gain, wherein said plurality of parameters further comprise codebook gain, wherein the processor performs the plurality of decoding steps to generate a codebook vector, wherein said processor scales said codebook vector by said codebook gain to generate a scaled codebook vector, wherein said processor generates a power signal representing the power of said scaled codebook vector, wherein said processor is

responsive to said pitch gain and said power signal to generate said adjusted first parameter, and wherein said adjusted first parameter comprises an adjusted pitch gain *(multiplying a vector by an optimal codebook gain to determine a power signal for error determination which is used to calculate an optimal pitch gain, Col. 7, Line 33- Col. 8, Line 21; and Col. 8, Lines 46-65).*

13. **Claims 7-8 and 40-41** are rejected under 35 U.S.C. 103(a) as being unpatentable over Yue et al in view of Ertem et al *(U.S. Patent: 6,453,289).*

With respect to **Claims 7 and 40**, Yue teaches the system for reducing noise in compressed speech data by substituting adjusted speech coefficients, as applied to Claims 1 and 34. Yue does not specifically suggest adjusting speech coefficients based upon pitch gain and signal to noise ratio (SNR), however Ertem discloses a means for selecting a pitch gain correction factor based upon a pitch gain and an estimated SNR *(Col. 10, Lines 19-45).*

Yue and Ertem are analogous art because they are from a similar field of endeavor in noise reduction in speech coding. Thus, it would have been obvious to a person of ordinary skill in the art, at the time of invention, to modify the teachings of Yue with the gain correction factor selection means disclosed by Ertem in order to achieve reliable noise estimation for noise reduction processing *(Ertem, Col. 1, Lines 38-44).*

With respect to **Claims 8 and 41**, Ertem further discloses:

The signal to noise ratio comprises a ratio involving noisy signal power and noise power of the audio signal *(SNR comprising speech and noise power levels, Col. 10, Lines 19-45).*

14. **Claims 10-12, 19, 43-45, and 52** are rejected under 35 U.S.C. 103(a) as being unpatentable over Yue et al in view of Chen *(U.S. Patent: 5,615,298)*.

With respect to **Claims 10 and 43**, Yue teaches the system for reducing noise in compressed speech data by substituting adjusted speech coefficients, as applied to Claims 1 and 34. Yue does not specifically suggest the use of a pitch gain at a buffer as recited in claims 10 and 43, however Chen discloses a process for determining pitch weighting for a first lag *(Col. 28, Line 11- Col. 29, Line 18)*.

Yue and Chen are analogous art because they are from a similar field of endeavor in noise reduction in speech coding. Thus, it would have been obvious to a person of ordinary skill in the art, at the time of invention, to modify the teachings of Yue with the process for determining pitch weighting for a first lag as taught by Chen in order to achieve pitch processing which ensures that voiced regions do not get amplified relative to unvoiced regions *(Chen, Col. 29, Lines 11-14)*.

With respect to **Claims 11 and 44**, Chen discloses a process for determining pitch weighting for a second lag *(Col. 28, Line 11- Col. 29, Line 18)*.

With respect to **Claims 12 and 45**, Chen discloses a long-term predictor buffer utilized for the first and second pitch lags *(Col. 28, Line 11- Col. 29, Line 18)*.

With respect to **Claims 19 and 52**, Chen further discloses the use of a long-term predictor code *(Col. 11, Lines 26-33)*.

15. **Claims 29-33 and 62-66** are rejected under 35 U.S.C. 103(a) as being unpatentable over Yue et al in view of the Applicants' Admitted Prior Art *(AAPA)*.

With respect to **Claims 29 and 62**, Yue discloses:

A processor responsive to the second bits to adjust the first bits and second bits, without decoding said compression code, whereby the noise characteristic in the digital signals is controlled *(replacing LPC parameters for noise conditioning in response to analyzing an excitation bit segment at a speech detector, Col. 4, Lines 1-7; and Col. 5, Line 66- Col. 8, Line 11)*; and

A transmitter module to transmit adjusted first and second bits to a device to produce a corresponding audible signal with managed noise for an end user *(signal processor, 100, located in a network element featuring a transmitter for sending the noise adjusted speech data, Fig. 3).*

Although Yue discloses adjusting bits directed to LPC compression code and further teaches a TFO GSM implementation *(Col. 1, Lines 18-32 and Col. 3, Lines 9-19)*, Yue does not explicitly récite the combination of a compression code and a linear code to express a speech signal, known to such a TFO GSM coding scheme, however, is well known in the prior art as is evidenced by the AAPA. The AAPA recites a TFO GSM standard using a combination of coded speech and PCM bits *(Page 2, Line 11- Page 3, Line 11; and Fig. 3).*

Yue and the AAPA are analogous art because they are from a similar field of endeavor in speech compression. Thus, it would have been obvious to a person of ordinary skill in the art, at the time of invention, to modify the teachings of Yue with the TFO GSM standard recited in the AAPA in order allow Yue's noise controller to comply with well-known cellular network standards *(AAPA, Page 2, Lines 17-19).*

With respect to **Claims 30 and 63**, the AAPA further recites the use of PCM code *(Page 2, Line 7- Page 3, Line 11).*

With respect to **Claims 31 and 64**, the AAPA recites the TFO GSM standard as applied to Claims 26 and 57.

With respect to **Claims 32 and 65**, the AAPA further recites first bits comprising the two LSBs and second bits comprising 6 MSBs *(Page 3, Lines 3-11; and Fig. 3).*

With respect to **Claims 33 and 66**, the AAPA further recites the use of PCM code for the 6 MSBs *(Page 2, Line 7- Page 3, Line 11; and Fig. 3).*

16. **Claims 67-68** are rejected under 35 U.S.C. 103(a) as being unpatentable over Yue et al in view of Malvar *(U.S. Patent: 6,029,126).*

With respect to **Claims 67-68**, Yue teaches the system for reducing noise in compressed speech data by substituting adjusted speech coefficients, as applied to Claims 1 and 34. Although Yue teaches parameter replacement as applied to claims 1 and 34, Yue does not specifically suggest that partial decoding generates noisy speech signal and, subsequently, an estimated clean speech signal. Malvar, however, discloses that signal modification can be performed on a received noisy speech signal in the compressed domain, wherein interfering signals *(i.e., noise)* would be suppressed, resulting in an estimated clean speech signal *(Col. 1, Line 50- Col. 2, Line 4; Col. 2, Lines 22-34; Col. 3, Lines 44-59).* Since this process is performed in the compressed domain and does not involve full decoding and re-encoding, it would inherently require partial decoding *(Col. 2, Lines 1-4).*

Yue and the Malvar are analogous art because they are from a similar field of endeavor in speech processing in the compressed domain. Thus, it would have been obvious to a person of ordinary skill in the art, at the time of invention, to modify the teachings of Yue with the

compressed domain signal enhancement concept taught by Malvar in order to provide a means of enhancing a speech in a noisy speech signal without adding further delay *(Malvar, Col. 2, Lines 1-4; and Col. 3, 44-59)*.

## *Conclusion*

17. The prior art made of record and not relied upon is considered pertinent to applicant's disclosure:

Bhaskaran *("Mediaprocessing in the Compressed Domain," 1996)*- discloses modification of audio data in the compressed domain, namely level adjustment *(see Section 3)*.

Levine et al *("A Sines + Transients + Noise Audio Representation for Data Compression and Time/Pitch Scale Modifications," 1998)*- discloses the concept of modifying a speech/audio signal in the compressed domain.

18. Any inquiry concerning this communication or earlier communications from the examiner should be directed to James S. Wozniak whose telephone number is (571) 272-7632. The examiner can normally be reached on M-Th, 7:30-5:00, F, 7:30-4, Off Alternate Fridays.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Patrick Edouard can be reached at (571) 272-7603. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free).

James S. Wozniak
10/4/2007